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Multipriority Framework Classes for Bandwidth Differentiation Service

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Abstract: Internet deploy new applications, such as voice over IP, video conference, video streaming, audio streaming, mission-critical financial data, employ the User Datagram Protocol (UDP) and have different requirements on the bandwidth from those of existing applications using Transmission Control Protocol (TCP). TCP-friendliness and multimedia-friendliness “contradict” with each other. The possible approaches to address these problems are to enhance the Internet with resource reservation, admission control and adopt source adaptation schemes such that the sending rate of an application is adjusted according to the current network condition. The source adaptation scheme, better preferred has 2 drawbacks i.e., it is more moderately responsive to transient changes in congestion, including a slower response to an abrupt change in the available bandwidth and the requirement of multimedia applications cannot be satisfied by equation-based source adaptation when a large number of flows share a link. The proposed system, develops a framework which extends the concepts of fairness, TCP-friendliness and TCP-compatibility, by designing TCP and UDP friendly source adaptation schemes and develop a less conservative framework to provide bandwidth differentiation service without any change to the router of the existing Internet. The proposed scheme is extended towards the handling of multiple bottleneck links available in the real time networks. In addition, the scheme combines with TCP slow-start, timeout to provide a relatively differentiated bandwidth service in the existing Internet, without any change to the router. The newly developed source adaptation scheme is improved to work with multiple priority classes.

Key words: Multipriority framework, User Data gram Protocol (UDP), Transmission Control Protocol (TCP), Choose Your Response Function (CYRF)

INTRODUCTION

Many new applications are being widely deployed in the Internet, such as voice over IP, video conference, video streaming, audio streaming, mission-critical financial data and so on. Many of these applications employ the user datagram protocol (UDP) and have different requirements on the bandwidth from those of existing applications using transmission control protocol (TCP). For example, multimedia applications have the following 2 distinguished requirements:

- The sending rate should be adjusted smoothly.
- The sending rate should be greater than a threshold.

TCP-friendly source adaptation schemes do not work well for multimedia applications. It seems that TCP-friendliness and multimedia-friendliness contradict with each other. Thus, these new applications present new problems for the Internet.

There are 2 possible approaches to address these problems. One is to enhance the Internet with resource reservation, admission control and so on. The other is to adopt source adaptation schemes such that the sending rate of an application is adjusted according to the current network condition. Compared to the first approach, the second one has the advantage of better utilization of the available time-varying network resources.

Literatrue review: Recently, an interesting framework, called Choose Your Response Function (CYRF), was proposed for memoryless window-based source adaptation protocols by using the fairness index.

Many existing congestion control schemes, like additive-increase multiplicative-decrease (AIMD), generalized AIMD (GAIMD) (Jain *et al*., 1984), binomial congestion control (BCC) (Bansal and Balakrishnana, 2001; Floyd *et al*., 2000) and are special cases of the CYRF. The CYRF has many nice features, such as: it converges to fairness and efficiency with a simple source adaptation scheme, it can be easily adapted to suit different applications and network requirements and it can be made TCP-friendly and so on. However, the CYRF has 2 disadvantages (Li *et al*., 2003; Wang and Schulxrinne, 2005).

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One is that it is difficult to use the CYRF to design multimedia friendly source adaptation schemes. The other is that the requirements of CYRF is still a little conservative. For example, a protocol named LOG, is not CYRF. Thus, the concept of CYRF is to cover the case, which is a protocol with a smooth increase policy whose and are monotonically non-decreasing (one of them must be strictly increasing) (Yang and Lam, 2000).

Another, important type of source adaptation is equation based source adaptation. The advantage of this type of source adaptation, as compared to the CYRF, is a smoother sending rate. It can also be used to maintain a relatively steady sending rate. However, it relies heavily on the accurate measurement of round trip time, steady-state loss event rate and the TCP retransmit time out value, especially the steady-state loss event rate that is very difficult to be measured (Jacobson and Karels, 1990). Furthermore, it has 2 disadvantages:

- It is more moderately responsive to transient changes in congestion, including a slower response to an abrupt change in the available bandwidth. As a result, it cannot aggressively find and use available bandwidth.
- The 2nd requirement of multimedia applications cannot be satisfied by equation-based source adaptation when a large number of flows share a link.

The proposed method, extends the concepts of fairness, TCP friendliness and TCP-compatibility, such that the following 2 objectives can be achieved.

- A platform can be set up to design TCP and UDP friendly source adaptation schemes.
- Bandwidth differentiation service can be provided by only using source adaptation with signal feed back from the destination.

This will increase the incentives for applications to use end-to-end source adaptations. Thus, contributing to the overall stability and evolution of the Internet. The proposed method, develops a less conservative framework to provide bandwidth differentiation service without any change to the router of the existing Internet.

SYSTEM MODEL

The system model of the thesis consists of the following components.

Monotonic response function: In this module a general framework for window-based memoryless source adaptation protocols that is convergent to the desired fairness is developed. In this framework a monotonically non-increasing function and a monotonically non-decreasing function and that the sending rates of flows are adjusted by increase policy and decrease policy.

Smoothness and efficiency: The smoothness of traffic adjustment is an important property when video, audio or speech is transmitted over the network. A window increase (decrease) policy is said to be smooth if the window size increase (decrease) from a single application is at least an order of magnitude smaller than the current window size.

Meanwhile, the policies are required to move the total bottleneck link utilization closer to the link capacity for efficient use of the bandwidth. This can be achieved by the principle of negative feedback, i.e., each flow increases its window size when the bottle link is under-utilized and decreases its window size when it is overloaded.

TCP fairness convergence: The system set 4 preconditions on the network.

Synchronization assumption: It is assumed that all the flows in the network get the same feedback and get the feedback simultaneously.

Feedback signal: The feedback is binary and limited to a single bit indicating whether the network is overloaded, i.e. a '1', or if the bandwidth is not fully utilized, i.e. a '0'.

Response functions: Assume that all sources use the same functions. When an ACK is received and the feedback is 1, the next window size is computed by the policy. If the feedback is 0, the policy is used to calculate the next window size.

Number of bottleneck links: The number of bottleneck links is satisfied after each application of policy or and at least one of the 2 policies must ensure a strict inequality.

TCP-Friendly MRF: To guarantee the display quality of video at the receiver side, smooth source adaptation schemes are desirable. This type of source adaptation schemes are also required to be TCP-friendly w.r.t. predefined weighting factors. The system shall derive necessary and sufficient condition for this MRF (Chen *et al*., 2007) with a smooth increase policy to be TCP-friendly w.r.t. the predefined weighting factor (Fig. 1).

It employed a new source adaptation agent instead of using the conventional TCP congestion avoidance mechanism to reduce the disturbance from

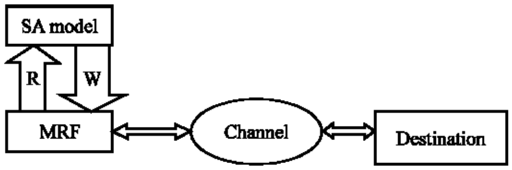

Fig. 1: Structure of source adaption model

other mechanisms such as slow-start or time-out etc. It has a single bottleneck link with a default bandwidth of 1 MB and a delay of 30 ms. The algorithm of RED is used at each router to drop packets in case of congestion. Several TCP-Reno flows (with different RTT) were introduced as the back ground traffic. The packet size is fixed at 500 bytes and all flows start at random times.

EXPERIMENTATION

The experimentation employed a new source adaptation agent instead of using the conventional TCP congestion avoidance mechanism to reduce the disturbance from other mechanisms such as slow-start or time-out etc. It has a single bottleneck link with a default bandwidth of 1 MB and a delay of 30 ms. The algorithm of RED is used at each router to drop packets in case of congestion. Several TCP-Reno flows (with different RTT) were introduced as the back ground traffic. The packet size is fixed at 500 bytes and all flows start at random times.

CONCLUSION

In the proposed method, differentiation bandwidth service can be provided by adopting the Monotonic Responsive Function (MRF) as the source adaptation scheme with necessary feedback signal from the destination. With the MRF as the source adaptation scheme, a flow can not only find and use the available bandwidth but also adjust its sending rate smoothly. More applications over the Internet can then be supported.

The proposed scheme is easier to use the Taylor's expansion to study this MRF with a smooth increase policy. The MRF can be easily used to analyze existing network protocols and construct new network protocols. The MRF does not need to measure round trip time; steady-state loss event rate and the TCP retransmit time out value as required by equation-based source adaptation. The MRF is a good platform for source adaptation of TCP and UDP. It can also be used to design TCP-friendly and multimedia-friendly source adaptation schemes. This is very helpful for multimedia applications over the Internet.

The MRF framework with a new fairness index analyze memoryless window based congestion control protocols. A necessary and sufficient condition is also derived for stepwise convergence of our MRF to the fairness. The condition is used to construct increase-decrease policies. The MRF framework is used for constructing existing protocols, like additive increase multiplicative-decrease (AIMD), generalized AIMD and binomial congestion control.

The proposed system handles multiple bottleneck links, which normally present between the source and destination. It is thus, of practical and theoretical interests to derive equivalent results for the case of multiple bottleneck links. In addition the proposed source adaptation works well for multiple priority classes. With our framework, one can design a new pricing system for the Internet by fully utilizing the feature of Internet service.

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